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DETAILED ACTION

Introduction

1. This action is response to amendment filed on 06-20-2005. Claims 1-11 are pending.

Claim Rejections - 35 USC § 112

2. The following is a quotation of the first paragraph of 35 U.S.C. 112:

The specification shall contain a written description of the invention, and of the manner and process of making and using it, in such full, clear, concise, and exact terms as to enable any person skilled in the art to which it pertains, or with which it is most nearly connected, to make and use the same and shall set forth the best mode contemplated by the inventor of carrying out his invention.

3. Claims 1, 8 and 10 are rejected under 35 U.S.C. 112, first paragraph, as failing to comply with the enablement requirement. The claim(s) contains subject matter which was not described in the specification in such a way as to enable one skilled in the art to which it pertains, or with which it is most nearly connected, to make and/or use the invention. The claimed limitation " a controller that causes the processor to maximize a power measure of the combined audio signal, wherein the controller is arranged to limit a combined power gain measure of the processed audio signals to a predetermined value without measuring an energy transfer at each site where one respective audio source of the plurality of audio sources receives the input audio signals" (see specification page 3 lines 7-17) was not supported in the further detail in the specification nor in any claim.

The specification does not teach that limiting a combined power gain measure of the processed audio signals to a predetermined value without measuring an energy transfer as is recited in the claim (see specification page 3 lines 7-17).

Claim Rejections - 35 USC § 103

4. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

(a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negated by the manner in which the invention was made.

5. Claims 1-5 and 7-10 rejected under 35 U.S.C. 103(a) as being unpatentable over in view of Kaneda (US PAT. 5,208,864) in view of Kellermann (US PAT 5,602,962).

Consider claim 1, Kaneda teaches an audio arrangement for that utilizes an energy transfer function for delay compensation, said arrangement comprising:

a plurality (see fig.12, (51)) of audio sources generating a plurality of input audio signals;

a processor (53₁ –53_m) comprising a scaling means (equations 2 and 3) for weighting the plurality of input signals and deriving a plurality of a processed audio signals from the plurality of input audio signals without delay values (see col.8 lines 10-37); and

a combiner (see fig.12, 55)) that derives a combined audio signal from the plurality of processed audio signals;

controller (54) that causes the processor to maximize (by minimizing a power of the noise equal to 0 and then the speech signal becomes maximizing and see col.7 line 58-col.8 line 35) a power measure (equation 2 and 3) of the combined audio signal, wherein the controller (54) is arranged to limit a combined power gain measure of the processed audio signals to a predetermined value without measuring (U_1-U_m) an energy transfer at each site where one respective audio source of the plurality of audio sources receives the input audio signals (see col.7 line 43-col.9 line 24); but Kaneda does not clearly teach to limit a combined power gain measure of the processed audio signals to a predetermined value.

However, Kellermann teaches that controller controls the processing in order to maximize a power measure of the combined audio signal (see col.3 line 45-col.4 line 55), which means signal-to-noise ratio (SNR) of the sum signal x at the output of the adder device 5 is maximized and in that the controller are arranged for limiting a combined power gain measure (see col.2 lines 1-12) of the processed audio signals to a predetermined value, which means the noise component of the sum signal x at the output of adaptive filter device 6 is to a threshold or predetermined level $SNR = (\sigma_s)^2 / (\sigma_n)^2$, $0 < (\sigma_n)^2 \leq 1$.

Therefore, it would be obvious to one of ordinary skill in the art at the time invention was made to combine the teaching of Kellermann in to Kaneda to improve noise reduction of the microphone signals and improve the audibility of speech.

Consider claim 2, Kaneda discloses that audio processing arrangement wherein the processor includes a scaling means (53_1-53_m) for scaling the input audio signals

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with a scaling factor (equations 2 and 3) for obtaining the processed audio signal (4), said controller (54) includes a further scaling means($53_1 - 53_m$) for deriving a plurality of scaled combined audio signals (55) with a scaling factor (equations 2 and 3) corresponding to the scaling factor (equations 2 and 3) of the scaling means ($53_1 - 53_m$), and in that the controller (5) is arranged for maximizing(by minimizing a power of the noise equal to 0 and then the speech signal becomes maximizing and see col.7 line58-col.8 line 35) power measure (equations 2 and 3) of the combined (55) audio signal, and for limiting a combined (55) power gain measure of the processed audio signals by minimizing a difference between the input audio signals and the scaled combined audio signals corresponding to said audio signals (see col.7 line 43-col.8 line 55).

Consider claim 3, Kaneda teaches that the audio processing arrangement wherein the processor ($53_1 - 53_m$) includes a plurality of adjustable filters ($53_1 - 53_m$) for deriving the processed audio signal, in that the controller (5) includes a plurality of further adjustable filters ($53_1 - 53_m$) having a transfer function being the conjugate of the transfer function of the adjustable filters ($53_1 - 53_m$), said further adjustable filters ($53_1 - 53_m$) being arranged for deriving from the combined (55) audio signal filtered combined audio signals, and in that the controller is arranged for maximizing (by minimizing a power of the noise equal to 0 and then the speech signal becomes maximizing and see col.7 line58-col.8 line 35) the power measure of the combined audio signal, and for restricting a combined power gain measure (equations 2 and 3) of the processed audio signals to a predetermined value by controlling the transfer functions of the adjustable filters($53_1 - 53_m$) and the further adjustable filters($53_1 - 53_m$)

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in order to minimize a difference measure between the input audio signals and the filtered combined audio signal corresponding to say input audio signals (see col.7 line 43-col.9 line 24).

Consider claim 4, Kellermann teaches the audio processing arrangement comprises delay elements (see fig.1, #2) for compensating a delay difference of a common audio signal present in the input audio signals (see col.3 line 18-60).

Consider claims 5 and 7 Kaneda teaches the audio processing arrangement of the audio sources comprise a plurality of microphones (see fig.12, (51)), and in that the microphones are placed in a position such that their directionality patterns are substantially disjunct (51); and the audio processing arrangement of the audio sources comprise a plurality of microphones being placed in a linear array (see fig.12 (51) and col.7 line 43-col.8 line 55).

Consider claim 8 Kaneda teaches an audio signal processing arrangement that utilizes an energy transfer function for delay compensation, said arrangement comprising a plurality of inputs (see fig.12, (51))) for receiving input audio signals, processing means ($53_1 - 53_m$) for deriving processed audio signals including scaling means (see equation 2-3) for scaling the input audio signal without delay values, the audio processing arrangement comprising combining means (9) for deriving a combined audio signal from the processed audio signals (4), the audio processing arrangement comprises a control means (54) for controlling the processing means ($53_1 - 53_m$) in order to maximize (by minimizing a power of the noise equal to 0 and then the speech signal becomes maximizing and see col.7 line 58-col.8 line 35) a power measure

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of the combined audio signal, and in that the control means (54) are arranged for limiting a combined power gain measure of the processed audio signals to a value without measuring an energy transfer ($U_1, U_2 \dots U_m$) at each site where each respective one the plurality of audio sources receives the input audio signals (see fig.12 and see col.7 line 43-col.9 line 24); but Kaneda does not clearly teach to limit a combined power gain measure of the processed audio signals to a predetermined value.

However, Kellermann teaches that a control means for controlling the processing means in order to maximize a power measure of the combined audio signal (see col.3 line 45-col.4 line 55), which means signal-to-noise ratio (SNR) of the sum signal x at the out put of the adder device 5 is maximized and in that the control means are arranged for limiting a combined power gain measure (see col.2 lines 1-12) of the processed audio signals to a predetermined value, which means the noise component of the sum signal x at the out put of adaptive filter device 6 is to a threshold or predetermined level $SNR = (\sigma_s)^2 / (\sigma_n)^2$, $0 < (\sigma_n)^2 \leq 1$.

Therefore, it would obvious to one of ordinary skill in the art at the time invention was made to combine the teaching of Kellermann in to Kaneda to improve noise reduction of the microphone signals and improve the audibility of speech.

Consider claim 10, there is a method claim corresponding to apparatus claim 8. See previous apparatus claim 8 rejection.

Consider claim 9, Kaneda discloses that the audio signal processing arrangement the scaling means (see fig.12, $53_1 - 53_m$) scale the input audio signals with a scaling factor (see equation 2-3) for obtaining the processed audio signals, said control means

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(54) comprise further scaling means (53₁ – 53_m) for deriving a plurality of scaled combined audio signals (55) with a scaling factor (see equation 2-3) corresponding to the scaling factor of the scaling means (53₁ – 53_m), and in that the control means (5e) are arranged for maximizing (by minimizing a power of the noise equal to 0 and then the speech signal becomes maximizing and see col.7 line58-col.8 line 35) a power measure of the combined audio signal, and for limiting a combined (55) power gain measure of the processed audio signals by minimizing a difference between the input audio signals and the scaled combined audio signals (see fig.12 and see col.7 line 43-col.9 line 24).

Claim Rejections - 35 USC § 103

6. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

(a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negated by the manner in which the invention was made.

7. Claim 6 is rejected under 35 U.S.C. 103(a) as being unpatentable over Kaneda (US PAT. 5,208,864) as modified by Kellermann (US PAT. 5,602,962) as applied to claims 1-5 above, and further in view of Kaneda (US PAT 4,536,887).

Consider claim 6, Kaneda (864) and Kellermann do not teach clearly the audio processing arrangement of the microphones are placed around a center position at angles being equal to 360 degrees divided by the number of microphones.

However, Kaneda (887) discloses that the audio processing arrangement includes that the microphones are placed around a center position at angles being

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equal to 360 degrees divided by the number of microphones (see fig.21d and col.20 line 10-col.21 line 20).

Therefore, it would obvious to one of ordinary skill in the art at the time invention was made to combine the teaching of Kaneda (887) into teaching of Kaneda (864) and Kellermann to provide microphone-array apparatus which can be constructed on a small scale and permits adaptive selection of the desired signal for varied positions of a desired signal and noise sources.

8. Claim 11 is rejected under 35 U.S.C. 103(a) as being unpatentable over in view of Kaneda (US PAT. 5,208,864) as modified by Kellermann (US PAT. 5,602,962) as applied to claim 10 above, and further in view of Anderson (US PAT 6,137,887).

Consider claim 11, Kaneda teaches the audio processing arrangement of further comprising a plurality of microphones having disjunct directionality patterns, wherein the audio signals are obtained from said plurality of microphones (see fig.12 (51)), but Kaneda fails to teach the microphone receiving a strongest speech signal is automatically emphasized.

However, Anderson teaches the microphone receiving a strongest speech signal is automatically emphasized ((greater than 9.5 db), see fig.3c and col.9 lines 5-53).

Therefore, it would obvious to one of ordinary skill in the art at the time invention was made to combine the teaching of Anderson into the teaching of Kaneda and Kellermann to provide an improved multiple-microphone audio system that identifies which microphone of a plurality of microphones best detects an audio source.

Response to Arguments

9. Applicant's arguments with respect to claims 1-11 have been considered but are moot in view of the new ground(s) of rejection.

Conclusion

10. The prior art made of record and not relied upon is considered pertinent to applicant's disclosure. Crow (US PAT. 5,657,393) is recited to show how other related the audio processing arrangement with multiple sources.

11. Any response to this action should be mailed to:

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Any inquiry concerning this communication or earlier communications from the examiner should be directed to Lao,Lun-See whose telephone number is (571) 272-7501. The examiner can normally be reached on Monday-Friday from 8:00 to 5:30.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Chin Vivian, can be reached on (571) 272-7848.

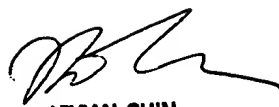
Any inquiry of a general nature or relating to the status of this application or proceeding should be directed to the Technology Center 2600 whose telephone number is (571) 272-2600.

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Date 08-30-2005



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